implementations does not have to be low-pass or high-pass. In some embodiments, this method may include using alternate filters, which may include, but are not limited to, a band-pass filter, to separate a signal into component waveforms.

[0086] There may be various benefits to using this post-processing method. These include, but are not limited to, ability to analyze and evaluate the filter EMD without concern with edge effects or filter warm up. In the following example, the Filter EMD has been applied to both an analytic signal:

$$s(k) = \sin\left(2\pi\frac{k}{911}\right) + \cos\left(2\pi\frac{k}{1001}\right) + \sin\left(2\pi\frac{t}{1493}\right) \qquad \text{EQN. 2}$$

[0087] and to an 18 second segment of recorded speech.

[0088] FIG. **19** and FIG. **20** show the measure of change, EQN. 1, for a sample, traditional and low-pass filter EMD methods as a function of the number of iterations of the Sifting Method performed. The high-pass filter EMD only used a single iteration. The data is shown for both IMFo and IMF **4**. The low-pass filter halts before than the traditional method, although not before the sample method, depending upon the value of the Cauchy criterion at which is chosen to stop. By comparing the number of extrema to number of zero-crossings, the sample method finishes first, while the low-pass filter EMD and the traditional method finish within one iteration of each other, FIG. **21**.

[0089] The situation is similar when the methods are applied to the acoustic signal. The high-pass filter EMD stops after a single iteration, followed by the sample method, the low-pass filter method and finally the traditional method. This is true whether the Cauchy criterion is used, FIG. **22**, or the extrema versus zero-crossing criteria, FIG. **23**.

[0090] The actual IMFs returned by the methods differ. FIG. **24** shows IMFo of the three sinusoid signal for all of the methods, while FIG. **25** shows the same for the acoustic signal. There is not an established criterion for judging the correctness of the EMD method. Since the EMD does not use a set of orthogonal basis functions, there is no guarantee of uniqueness. This means that the correctness of the decomposition will have to be judged by each application. There may be many benefits to a filter-based EMD, including but not limited to, the particular characteristics pulled out of the function may be tailored by the characteristics of the filter used. In addition, the filter may be adapted from iteration to iteration to reinforce elements of interest and suppress items of no value.

[0091] The processor may be any processor known in the art. The speaker and microphone may be any speaker and microphone known in the art. In some embodiments, one or more hearing aid speakers are used.

[0092] While the principles of the invention have been described herein, it is to be understood by those skilled in the art that this description is made only by way of example and not as a limitation as to the scope of the invention. Other embodiments are contemplated within the scope of the present invention in addition to the exemplary embodiments shown and described herein. Modifications and substitutions by one of ordinary skill in the art are considered to be within the scope of the present invention.

What is claimed is:

1. A system for processing audio signals comprising:
a hearing aid speaker;
a hearing aid microphone; and
at least one processor configured to:
receive an audio signal from the hearing aid microphone;
perform initial processing comprising filtering the audio signal to remove audio features, the initial processing resulting in a modified audio signal;
identify a phoneme;
replace the phoneme; and
transmit the modified audio signal to the hearing aid speaker.

2. The system of claim **1**, wherein the at least one processor produces an audio stream.

3. The system of claim **1**, wherein the at least one processor is configured to filter noise.

4. The system of claim **1**, wherein the at least one processor is configured to remove the audio features.

5. The system of claim **1**, wherein the at least one processor is configured to monitor the audio signal and any background noise to provide feedback.

6. The system of claim **1**, wherein the at least one processor is configured to classify the phoneme to enhance accuracy of phoneme identification.

7. The system of claim **1**, wherein the at least one processor is configured to:
monitor the audio signal and any background noise; and
provide feedback;
enhance the audio signal; and
provide information for classification.

8. The system of claim **1**, wherein the at least one processor is configured for phoneme identification by analyzing the modified audio signal using a Hilbert-Huang transform method.

9. The system of claim **1**, wherein the at least one processor is configured to identify a time slot occupied by the phoneme and identify the phoneme in the modified audio signal.

10. The system of claim **9**, wherein the at least one processor is configured to transmit the time slot and the identified phoneme for phoneme replacement.

11. The system of claim **10**, wherein the at least one processor is configured to determine whether the identified phoneme in an audio stream is a replaceable phoneme and replace the identified phoneme in the modified audio signal with a replacement signal.

12. The system of claim **11**, wherein the at least one processor is configured to receive the replacement signal from a table and determine a way to smoothly incorporate the replacement signal into the modified audio signal.

13. The system of claim **12**, wherein the at least one processor is configured to replace the identified phoneme to transmit the modified audio signal to the hearing aid speaker.

14. The system of claim **1**, wherein the at least one processor is configured to digitally filter extreme values of the audio signal.

15. The system of claim **1**, wherein the at least one processor is configured to:
process the audio signal and finding a maxima and a minima of the audio signal;
pass the maxima to a high-pass filter;
filter the maxima using a high pass filter to produce a filtered audio signal;
sample the filtered audio signal;